**In the claims:**

Please amend the claims as follows:

Please cancel claims 10, 11, 13, and 19.

1(currently amended). A telephony communications network comprising:

a telephony unit generating a private telephony signaling code, the telephony unit is a Session Initiation Protocol (SIP) user agent;

a translator coupled to the telephony unit, the translator encapsulating the private telephony signaling code in a application layer control protocol message, the application layer control protocol is a SIP; and

a communications interface coupled to the translator for transmitting the message over a communications network.

2(previously presented). The network of claim 1 further comprising:

one or more second telephony units; and

one or more second translators coupled to the one or more second telephony units, characterized in that the one or more second translators receive the message transmitted over the communications network and decapsulate the private telephony signaling code in the message, further characterized in that the one or more second translators forward the private telephony signaling code to the one or more second telephony units for performing a function in response to the private telephony signaling code.

3(original). The network of claim 2 further comprising a server for routing the message to the one or more second translators.

4(original). The network of claim 3, wherein the server provides a third party service for the telephony unit.

5(original). The network of claim 2, wherein the translator determines the one or more second translators viable for receiving the message and transmitting the message to the viable translators.

6(original). The network of claim 2, wherein the one or more second translators subscribe with the translator for receiving the message.

7(original). The network of claim 2, wherein the one or more second translators receive the message in an out-of-call data transfer.

8(original). The network of claim 1, wherein the telephony unit is a digital telephone.

9(original). The network of claim 1, wherein the telephony unit is a private branch exchange unit.

10-11(canceled).

12 (currently amended). A telephony communications network supporting a session initiation protocol (SIP) session, the network comprising:

a SIP client transmitting and receiving SIP messages during the SIP session; and

a translator coupled to the SIP client, the translator being an application programming interface, the translator configured to encapsulate and decapsulate private telephony signaling codes in and from the SIP messages for allowing the SIP client access to PBX functionality associated with the private telephony signaling codes.

13-15 (canceled).

16(previously presented). In a telephony communications network comprising a telephony unit and a translator coupled to the telephony unit, the translator comprising:

a signaling interface receiving a private telephony signaling code;

a processor coupled to the signaling interface, the processor configured to encapsulate the private telephony signaling code in a session layer control protocol message, the session layer control protocol being a Session Initiation Protocol (SIP);

the telephony unit being a SIP user agent; and

a network interface coupled to the processor for transmitting the message over a communications network.

17(previously presented). The translator of claim 16, wherein the processor is further configured to decapsulate a second private telephony signaling code in a second session layer control protocol message received by the network interface for forwarding to the telephony unit over the signaling interface.

18-19 (canceled).

20(original). The translator of claim 16, wherein the telephony unit is a digital telephone.

21(original). The translator of claim 16, wherein the telephony unit is a private branch exchange unit.

22-28 (canceled)